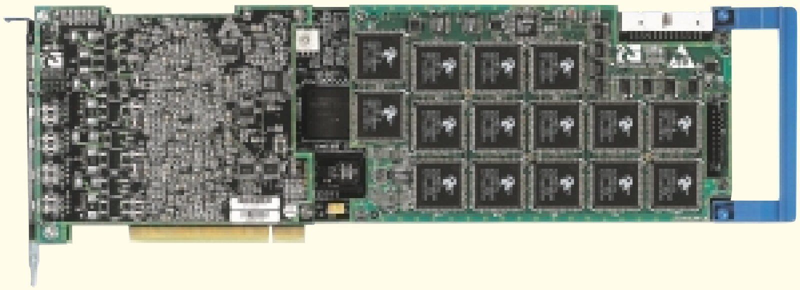
Enterprise VoIP Solutions Data Sheet

AudioCoded™ TP-240

PCI VoIP Communication Board



Features

- 128 voice/fax/data ports
- VoIP packet streaming (RTP/RTCP)
- Single slot plug and play PCI board
- On-board 10/100Base-T Network interface
- MVIP and SCbus TDM interfaces
- Integrated Single/Dual/quad E1/T1 interfaces
- Media Gateway on a blade mode
- MGCP (RFC2705) enabled
- MEGACO (H.248) enabled
- Management Interfaces: SNMP, Embedded Web Server

Applications

- Trunking Gateways
- IP-IVR
- IP-enabled Call Centers
- IP-PBXs
- Unified Messaging and Voice Mail

Overview

The AudioCoded™ TrunkPack® **(TP-240)** is an ideal solution for trunking gateways to the PSTN and integrated gateways for: IP-PBXs, all-in-one communication servers, unified messaging servers, voice mail servers and IVR (Interactive Voice Response Units). The **TP-240** provides 128 ports for voice, fax or data implementing VoIP media gateway applications. Powered by AudioCodes' VoIP DSP chips and advanced voice processing technologies, the **TP-240** offers excellent voice quality utilizing the superior G.168 compliant echo cancellation, lost packet interpolation and jitter buffer control. Additional features include G.723.1, G.729A, GSM-FR and proprietary NetCoder® voice compression. The board supports G3 fax relay/bypass, modem relay/bypass, silence suppression, DTMF detection/generation.

The **TP-240** features the MVIP and SCbus TDM data interfaces for integration with other 3rd party CTI boards. An on-board NIC interfaces the IP network for control and/or standard RTP/RTCP media streaming protocols. An optional Single/Dual/quad E1/T1 interface daughter-board provides TP-240 users a higher level of integration, saving backplane slot space, enabling higher density gateway platforms while reducing the costs per channel.

A dedicated communication processor implements the RTP/RTCP packet streaming, offloading the packets directly to and from the on-board NIC. This architecture enhances the scalability of the solution by freeing the host processor to perform call control protocols such as MEGACO, MGCP and application management. A sophisticated dynamic jitter buffer enables protection against the effects of network delay jitter.

The **TP-240** is an ideal solution for transporting voice and fax traffic over IP packet networks, allowing smooth migration from enterprise legacy systems to the new VoIP infrastructure. Supported by the same API as the rest of the TrunkPack® family, the **TP-240** protects the investment of AudioCodes' customers and leverages existing gateways by providing additional resources for new features and concurrent applications.

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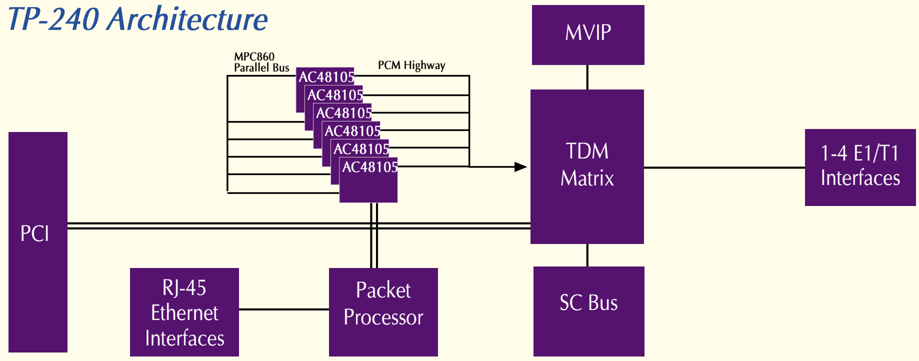
Selected Specifications

TP-240 Architecture



Capacity
30, 60, 128 independent digital voice, fax and data ports

Voice Compression
- G.723.1 P-MLQ at 6.3 kbps
- G.729A CS-ACELP at 8 kbps
- G.711 PCM at 64 Kbps (of µ/A-law)
- G.726/G.727 ADPCM / E-ADPCM at 16-40 Kbps
- NetCoder® Proprietary at 6.4-9.6 kbps, 800 bit steps
- GSM 6.10 at 13 kbps

Echo Cancellation
- G.168, 25 msec

Silence Suppression
- G.723.1 Annex A
- G.729 Annex B
- PCM & ADPCM - Proprietary VAD & CNG
- GSM 6.10 at 13 kbps
- NetCoder®

Gain Control
Programmable

Fax Relay
- Group 3 real time fax relay up to 14.4 kbps with auto fallback /Tolerant , delay up to 9 seconds
- T.38 compliant

Modem Transparency
Auto-switch to PCM or ADPCM on V.34 / V.90 modem detection

VoIP Standards Compliance
- DTMF over RTP per RFC 2833
- RTP/RTCP per RFC1889/1890

Control Protocols
- TPNCP - AudioCodes' proprietary VoIP API Library
- Media Gateway on a blade mode:
  - controlled by either MGCP or MEGACO
  - PCI used for power only

Management Interfaces
- SNMP V2 : Standard MIB-2 , RTP MIB, Trunk MIB , AudioCodes' proprietary MIB
- Embedded Web Server

In-band Signaling
- DTMF (TIA 464B)
- MFC-R2
- MFR1
- Call Progress Tones generation and detection

PSTN Protocol Termination:
- CAS
  - T1 robbed bit: WinkStart, delay dial, immediate start, FGB, FGD...
  - MFC/R2 numerous country variants
  - Unique script for each county variant, enabling maximum flexibility of the entire state machine of each CAS protocol.
- CCS

  ISDN PRI: ETSI EURO ISDN, ANSI NI2, DMS, 5ESS, Japan INS1500, QSIG Basic Call, Australian Telecom, New Zealand Telecom, Hong Kong Variant, Korean MIC

Telephony Interfaces
1, 2 or 4 Trunks of 4 E1 or T1 trunks 120 Ohm- RJ48C connectors (75 Ohm= with external 3rd party BNC/RJ- 48C adapter cables)

TDM Interfaces
- SCbus, 1024 time-slots at 4.096 MHz
- MVIP bus, 512 time-slots at 2.048 MHz

Ethernet
10/100 Base-T (Auto Negotiation)

Control Control Processors
Motorola PowerQUICC MPC860

Core DSP
AudioCodes' AC48105-C VoIP DSP

Power
- 3.0A at 5 V without E1/T1 interface
- 3.6A at 5 V with quad E1/T1 interface

Physical
Full length 32bit PCI, Rev 2., slave, 33 MHz

Operating System
- Windows NT 4.0, Win-NT library
- Linux
- Solaris / INTEL
- Solaris / SPARC

Ordering Information

TP-240 - SC-Bus I/F + MVIP I/F + 100BT + 120 Ohms

Device Name	Channels	Configuration
TP-240A-030XXDFU-R	30	No Trunks I/F
TP-240A-0301TDFU-R	30	1 E1/T1 Trunk I/F
TP-240A-060XXDFU-R	60	No Trunks I/F
TP-240A-0602TDFU-R	60	2 E1/T1 Trunk I/F
TP-240A-120XXDFU-R	120	No Trunks I/F
TP-240A-1204TDFU-R	120	4 E1/T1 Trunk I/F

* R - Designates different vocoder options

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